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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary

Application No.

10/561,961

Applicant(s)

TOWNS-VON STAUBER ET AL.

Examiner

CANDAL ELPENORD

Art Unit

2616

Period for Reply -- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

- 1) ☒ Responsive to communication(s) filed on 22 December 2005.
- 2a) ☐ This action is **FINAL**. 2b) ☒ This action is non-final.
- 3) ☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

- 4) ☒ Claim(s) 1-64 is/are pending in the application.
- 4a) Of the above claim(s) _____ is/are withdrawn from consideration.
- 5) ☐ Claim(s) _____ is/are allowed.
- 6) ☒ Claim(s) 1-49 and 52-64 is/are rejected.
- 7) ☒ Claim(s) 50-51 is/are objected to.
- 8) ☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

- 9) ☒ The specification is objected to by the Examiner.
- 10) ☒ The drawing(s) filed on _____ is/are: a) ☐ accepted or b) ☒ objected to by the Examiner.
- Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).
- Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).
- 11) ☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

- 12) ☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
- a) ☒ All b) ☐ Some * c) ☐ None of:
1. ☐ Certified copies of the priority documents have been received.
 2. ☐ Certified copies of the priority documents have been received in Application No. _____.
 3. ☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

- 1) ☒ Notice of References Cited (PTO-892)
- 2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
- 3) ☒ Information Disclosure Statement(s) (PTO/SF-08)
- Paper No(s)/Mail Date 22 December 2005
- 4) ☐ Interview Summary (PTO-413)
- Paper No(s)/Mail Date _____
- 5) ☐ Notice of Inventor's Patent Application
- 6) ☐ Other: _____

DETAILED ACTION

Drawings

1. The drawings are objected to because the descriptive legends in fig. 7 to fig. 13 are not legible. Corrected drawing sheets in compliance with 37 CFR 1.121(d) are required in reply to the Office action to avoid abandonment of the application. Any amended replacement drawing sheet should include all of the figures appearing on the immediate prior version of the sheet, even if only one figure is being amended. The figure or figure number of an amended drawing should not be labeled as "amended." If a drawing figure is to be canceled, the appropriate figure must be removed from the replacement sheet, and where necessary, the remaining figures must be renumbered and appropriate changes made to the brief description of the several views of the drawings for consistency. Additional replacement sheets may be necessary to show the renumbering of the remaining figures. Each drawing sheet submitted after the filing date of an application must be labeled in the top margin as either "Replacement Sheet" or "New Sheet" pursuant to 37 CFR 1.121(d). If the changes are not accepted by the examiner, the applicant will be notified and informed of any required corrective action in the next Office action. The objection to the drawings will not be held in abeyance.
2. In addition to Replacement Sheets containing the corrected drawing figure(s), applicant is required to submit a marked-up copy of each Replacement Sheet including annotations indicating the changes made to the previous version. The marked-up copy must be clearly labeled as "Annotated Sheets" and must be

presented in the amendment or remarks section that explains the change(s) to the drawings. See 37 CFR 1.121(d)(1). Failure to timely submit the proposed drawing and marked-up copy will result in the abandonment of the application.

Claim Rejections - 35 USC § 101

3. 35 U.S.C. 101 reads as follows:

Whoever invents or discovers any new and useful process, machine, manufacture, or composition of matter, or any new and useful improvement thereof, may obtain a patent therefor, subject to the conditions and requirements of this title.

Claims 54-64 are rejected under 35 U.S.C. 101 because the claimed invention is directed to non-statutory subject matter.

Regarding claims 54, the limitation “computer readable storage medium storing instructions” seems to refer back to “a suitable electronics communication medium” recited in page 35 in the specification. To overcome this rejection, it is suggested to applicant to remove the phrase “suitable electronics communication medium”. Similar problem exist in claim 60.

Claims 55-59, 61-64 are rejected since they depend on claims 54, 60.

Claim Rejections - 35 USC § 112

4. The following is a quotation of the second paragraph of 35 U.S.C. 112:

The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

5. **Claims 15-16** are rejected under 35 U.S.C. 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention.

Art Unit: 2616

Regarding claim 15, the limitation "K values are constant" are vague and indefinite because "K1, K2, K3, K4" are not properly specified in the claimed language. Additionally, the constant values of K need to be specified. Similar problem exists in claim 16.

Claim Rejections - 35 USC § 103

6. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

(a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negated by the manner in which the invention was made.

7. The factual inquiries set forth in *Graham v. John Deere Co.*, 383 U.S. 1, 148 USPQ 459 (1966), that are applied for establishing a background for determining obviousness under 35 U.S.C. 103(a) are summarized as follows:

1. Determining the scope and contents of the prior art.
2. Ascertaining the differences between the prior art and the claims at issue.
3. Resolving the level of ordinary skill in the pertinent art.
4. Considering objective evidence present in the application indicating obviousness or nonobviousness.

8. This application currently names joint inventors. In considering patentability of the claims under 35 U.S.C. 103(a), the examiner presumes that the subject matter of the various claims was commonly owned at the time any inventions covered therein were made absent any evidence to the contrary. Applicant is advised of the obligation under 37 CFR 1.56 to point out the inventor and invention dates of each claim that was not commonly owned at the time a later invention was made in order for the examiner to consider the applicability of

Art Unit: 2616

35 U.S.C. 103(c) and potential 35 U.S.C. 102(e), (f) or (g) prior art under 35 U.S.C. 103(a).

9. **Claims 1-9, 13-14, 31-39, 43-44, 54, 59-60, 62**, are rejected under 35 U.S.C. 103(a) as being unpatentable over Houh et al (US 2002/0015387 A1) in view of McLampy et al (US 7,151,781 B2).

Regarding claims 1, 31, 54, 60, Houh et al. discloses a method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9), comprising: passively sampling packets (fig. 5, Packets and Analysis Tool 170, "passively analysis on a pair of ports", recited in paragraph 0060) from a stream of Internet Protocol (IP) packets ("RTP Streams", recited in paragraph 0060, lines 6-16) that represent a communication session("RTP streams between the gateways", recited in paragraph 00014, lines 1-7, "packetized switched applications such as streaming video, video conferencing" recited in paragraph 0037, lines 5-8) between a pair of end points (fig. 5, VoIP Gateways 130, 150, recited in paragraph 0044) carrying analog signals ("packetized switched applications such as streaming video, video conferencing" recited in paragraph 0037, lines 5-8) being transmitted over a transmission path (fig. 5, Transmission path between the VoIP gateways) in an IP network (fig. 2, Packet Network 140, recited in paragraph 0040), and determining ("monitor of a group of RTP streams to determine packet jitter, packet loss", recited in paragraph 0065, lines 1-7) in near real time at least two metrics from the sampled packets (fig. 5, Packet Captures and Analysis Tool 170, "monitors of RTP streams for packet

lost, jitter", recited in paragraph 0060, lines 13-19) for the communication session ("RTP streams between the gateways", recited in paragraph 00014, lines 1-7); wherein the at least two metrics ("packet loss, packet jitter", recited in paragraph 0060, lines 13-19) comprise: at least one metric that measures a quantity of lost packets ("number of packet lost across all streams", recited in paragraph 0061, lines 6-12); and at least one metric that measures a characteristic of packet timing ("packets delay", recited in paragraph 0046, lines 6-12);

Regarding claims 3, 33, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9) according to claim 31, wherein the packets are sampled at a switch (fig. 1, Switch Fabric 35, recited in paragraph 0031, lines 1-6).

Regarding claims 4, 34, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9) according to claim 31, wherein the samples are taken at a switch (fig. 1, Switch Fabric 35, recited in paragraph 0031, lines 1-6)., and wherein the samples are taken of all packets entering and leaving the switch (fig. 5, see, Network Packet Capture and Analysis Tool 170, analyzes of packet s at the flow level between the connecting gateways, recited in paragraph 0060, lines 7-17).

Regarding claims 5, 35, the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9, fig. 5, Packet

Capture and Passive Analysis Tool 170, recited in paragraph 0060, lines 1-11) further comprising receives additional metrics ("voice data provided to the test unit 180 to account for delay, jitter, packet loss", recited in paragraph 0047, line 10-15) from network devices ("VoIP gateways 130, 150", recited in paragraph 0044) residing in the IP network (fig. 2, Packet Network 140, recited in paragraph 0040) along the transmission path (fig. 5, Transmission path between the VoIP gateways).

Regarding claims 6, 36, the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9) , the additional metrics comprise at least one of a soft switch call metric ("soft switch controls the call set-up on the data side of the gateway", recited in paragraph 0036, lines 6-12), call metrics stored on an end-device, a VoIP (Voice over IP) network component (fig. 3, VOIP Gateway 150, recited in paragraph 0044), and a Network Performance Test Probe (NPTP) result (fig. 1, Test Unit 180, recited in paragraph 0046, lines 13-30).

Regarding claims 7, 37, the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9) according to claim 31, wherein the Internet Protocol (IP) packets that represent analog voice signals ("voice data", recited in paragraph 4-7) in which digitized voice is carried in Real Time Protocol (RTP) packets ("packets received form RTP streams", recited in paragraph 0015, lines 9-12).

Regarding claims 8, 38, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines

1-9) according to claim 31, wherein the at least two metrics ("packet loss, packet jitter", recited in paragraph 0014).

Regarding claims 9, 39, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9).

Regarding claims 13, 43, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9).

Regarding claims 14, 44, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9), wherein the process (fig. 1, Network Processor 1, recited in paragraphs 0027-0028) is carried out on a programmed processor (fig. 1, Programmed Channel Processor 10, recited in paragraph 0029, lines 1-6).

Houh et al. discloses all the claimed limitation with the exception of being silent with respect to the following claimed features: **regarding claims 1, 31, 54, 60,** and calculating a quality score in near real time using a quality formula that combines the at least two metrics; **regarding claims 5, 35,** and wherein the quality formula incorporates functions of the additional metrics; **regarding claims 8, 38,** are derived from data in Real Time Control Protocol (RTCP) packets; **regarding claims 9, 39,** further comprising storing the quality score in a database indexed to the pair of end points; **regarding claims 13, 43, 59** further comprising aggregating a plurality of quality scores.

However, McLampy et al. in the same field of endeavor discloses the above claimed features: **regarding claims 1, 31, 54, 60**, and calculating a quality score ("QoS score dynamically determined", recited in col. 13, lines 64 – col. 14, lines 12) in near real time using a quality formula that combines ("QoS is computed that summarizes the three statistics into a single value", recited in col. 17, lines 23-32) the at least two metrics ("packet loss, packet latency, jitter", recited in col. 16, lines 66 – col. 17, lines 12); **regarding claims 5, 35**, and wherein the quality formula incorporates functions of the additional metrics; **regarding claims 8, 38**, are derived from data in Real Time Control Protocol (RTCP) packets ("real-time transport protocol packets", recited in col. 10, lines 20-27); **regarding claims 9, 39, 56, 62**, further comprising (fig. 3, Flow Quality Measurement Engine 348, recited in col. 12, lines 8-15, "determining jitter, packet loss", recited in col. 17, lines 1-12) storing the quality score ("QoS score stored in realm database", recited in col. 14, lines 5-12) in a database (fig. 4, QoS score column 346, "store a QoS score", recited in col. 13, lines 64 – col. 14, lines 3) indexed to the pair of end points (fig. 2, First Realm and Second Real 252, recited in col. 7, lines 61 – col. 8, lines 2); **regarding claims 13, 43, 59, 64** further comprising aggregating a plurality of quality scores ("QoS is computed that summarizes the three statistics into a single value", recited in col. 17, lines 23-32). Therefore, it would have been obvious to one skill in the art at the time the invention was made to modify the features of Houh et al. by using features as taught by McLampy et al. in order to provide session admission control and quality of service (See Col. 7, lines 16-24 for motivation).

10. **Claims 2, 10-12, 32, 40-42, 55, 57-58, 61, 63** are rejected under 35 U.S.C. 103(a) as being unpatentable over Houh et al. in view of MeLampy et al (US 7,151,781 B2) as applied to claims 1, 31, 54, 60 above, and further in view of Mitsumori et al (US 6,850,525 B2).

Regarding claims 2, 32, 55, 61, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9), wherein the at least one metric that measures a characteristic of packet timing measures at least one of packet jitter ("packet jitter", recited in paragraph 0015, lines 1-9), packet latency ("packet latency", recited in paragraph 0015, lines 1-9).

Regarding claims 10, 40, 63, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9).

Regarding claims 11, 41, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9).

Regarding claims 12, 42, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9).

Houh et al. and MeLampy et al. disclose all the claimed limitation with the exception of being silent with regard to the following claimed features: **regarding**

claims 2, 55, 61, round trip time; **regarding claims 10, 40, 57, 63**, further comprising generating an alarm whenever the quality score falls below a quality threshold; **regarding claims 11, 41, 58**, further comprising displaying the quality score on a display; **regarding claims 12, 42**, wherein the display further displays historical quality scores associated with the end points.

Mitsumori et al (US 6,850,525 B2) in the same field of endeavor discloses the above claimed features: **regarding claims 2, 32, 55, 61**, round trip time ("measures of round trip delay", recited in col. 1, lines 66 – col. 2, lines 10); **regarding claims 10, 40, 57, 63**, further comprising generating an alarm ("monitoring device to trigger an alarm", recited in col. 2, lines 49-60) whenever the quality score falls below a quality threshold ("network performance statistics exceeds a preconfigured threshold", recited in col. 2, lines 49-60); **regarding claims 11, 41, 58**, further comprising displaying (fig. 3, "graphical User Interface", recited in col. 6, lines 15-35) the quality score on a display; **regarding claims 12, 42**, wherein the display (fig. 3, Web Server 116, "the web server may be configured to display", recited in col. 6, lines 1-26) further displays historical quality scores ("histograms that show distribution of packet loss, jitter, round-trip delay", recited in col. 5, lines 48-62) associated with the end points (fig. 1, VoIP Points of Presence 104a, 104b, recited in col. 3, lines 37-59). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the features of Houh et al. with McLampy et al. by using features as taught by Mitsumori et al. in order to provide in order to generate network performance statistics for VoIP network (See Col. 1, lines 43-65 for motivation).

10. **Claims 17-27, 45-49** are rejected under 35 U.S.C. 103(a) as being unpatentable over Houh et al (US 2002/0015387 A1) in view of McLampy et al (US 7,151,781 B2) as applied to claims 1, 31 above, and further in view of Mitsumori et al (US 6,850,525 B2), Baj et al (US 2002/0145979 A1), Hussain et al (US 2003/0223361 A1).

Regarding claims 17, 45, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9).

Regarding claim 18, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9).

Regarding claims 19-23, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9).

Regarding claims 24, 47, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9).

Regarding claims 25, 48, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9).

Regarding claims 26, 49, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9).

Regarding claim 27, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9).

Houh et al. and McLampy et al. disclose all the claimed limitation with the exception of being silent with regard to the following claimed features: **regarding claim 18**, wherein the quality formula is designed to approximate a Perceptual Speech Quality Measurement (PSQM) score; **regarding claim 19**, wherein an alert threshold level is defined for quality scores of approximately 3.5; **regarding claim 20**, wherein an alert threshold level is defined for quality scores of approximately 3.3 to 3.7; **regarding claim 21**, wherein an alert threshold level is defined for quality scores of approximately 2.8; **regarding claim 22**, wherein an alert threshold level is defined for quality scores of approximately 2.5 to 3.1; **regarding claim 23**, wherein a low level alert threshold level is defined for quality scores of approximately 2.8, and wherein a higher level alert threshold level is defined for quality scores of approximately 3.5, **regarding claim 46**, wherein the quality formula is designed to approximate a Perceptual Speech Quality Measurement (PSQM) score, and wherein a low level alert threshold level is defined for quality scores of approximately 2.8, and wherein a higher level alert threshold level is defined for quality scores of approximately 3.5.

Mitsumori et al. in the same field of endeavor discloses the above claimed features: **regarding claims 19-22**, the alert threshold level ("triggering a

traceroute between the first VoIP and the second VoIP when the network performance statistics exceeds a preconfigured threshold", recited in col. 2, lines 11-20), **regarding claims 23-25**, low, and high alert threshold (see, "these thresholds may be expressed in the format of SLA for VoIP services", recited in col. 4, lines 67 - col. 5, lines 14); wherein the quality scores exceeding the low level alert threshold but not the higher level alert threshold, wherein the quality scores exceeding the higher level alert threshold but not the higher level alert threshold, wherein a low level alert and a high level alert threshold are established.

Regarding claim 26, further comprising means for displaying (fig. 3, see "QoS Chart", recited in col. 6, lines 15-38) the quality score ("performance measurements with regard to jitter, packet lost, roundtrip delay", recited in col. 5, lines 48-62) on a display (fig. 3, see "QoS Chart", recited in col. 6, lines 15-38) to indicate the quality score's relationship ("performance measurements with regard to jitter, packet lost, roundtrip delay", recited in col. 5, lines 48-62) to the alert levels ("trigger an alarm for high packet loss, high jitter, high round0trip-time", recited in col. 4, lines 52-67).

Regarding claims 27, wherein the quality score ("performance measurements with regard to jitter, packet lost, roundtrip delay", recited in col. 5, lines 48-62) is compared to at least one threshold ("examines the average and maximum values with regard to samples of data", recited in col. 4, lines 32-57) that is established either manually or dynamically ("configured automatically", recited in col. 4, lines 21-42).

Regarding claim 46, low and higher level alert threshold ("preconfigured threshold", recited in col. 4, lines 52-67).

Regarding claim 47, wherein the quality scores exceeding the low level alert threshold ("trigger an alarm when the network performance statistics exceeds pre-configured threshold", recited in col. 2, lines 49-60) but not the higher level alert threshold, wherein the quality scores exceeding the higher level alert threshold but not the higher level alert threshold("trigger an alarm when the network performance statistics exceeds pre-configured threshold", recited in col. 2, lines 49-60).

Regarding claims 48, wherein a low level alert and a high level alert threshold are established ("preconfigured threshold", recited in col. 4, lines 52-67), wherein the quality scores exceeding the low level alert threshold but not the higher level alert threshold ("trigger an alarm when the network performance statistics exceeds pre-configured threshold", recited in col. 2, lines 49-60), wherein the quality scores exceeding the higher level alert threshold but not the higher level alert threshold, ("trigger an alarm when the network performance statistics exceeds pre-configured threshold", recited in col. 2, lines 49-60).

Regarding claim 49, wherein the quality score ("performance measurements with regard to jitter, packet lost, roundtrip delay", recited in col. 5, lines 48-62) is compared to at least one threshold ("examines the average and maximum values with regard to samples of data", recited in col. 4, lines 32-57) that is established either manually or dynamically ("configured automatically", recited in col. 4, lines 21-42). Therefore, it would have been obvious to one of

ordinary skill in the art at the time the invention was made to modify the features of Houh et al. with MeLampy et al. by using features as taught by Mitsumori et al. in order to generate network performance statistics for VoIP network (See Col. 1, lines 43-65 for motivation).

Houh, MeLampy, and Mitsumori et al. disclose all the claimed limitation with the exception of being silent with regard to the following claimed features: **regarding claim 18**, wherein the quality formula is designed to approximate a Perceptual Speech Quality Measurement (PSQM) score; **regarding claim 19**, quality scores of approximately 3.5, **regarding claim 20**, quality scores of approximately 3.3 to 3.7; **regarding claim 21**, quality scores of approximately 2.8; **regarding claim 22**, quality scores of approximately 2.5 to 3.1; **regarding claims 17, 45**, wherein the quality formula is developed by matching observed quality for communications over the IP network to a standard quality measurement scale, and equating the observed quality to the quality score as the at least two metrics are varied; **regarding claim 46**, the quality formula is designed to approximate a Perceptual Speech Quality Measurement (PSQM) score, and wherein a low level alert threshold level is defined for quality scores of approximately 2.8, and wherein a higher level alert threshold level is defined for quality scores of approximately 3.5.

Baj et al. in the same field of endeavor discloses the above claimed features: **regarding claim 18**, wherein the quality formula is designed to approximate a Perceptual Speech Quality Measurement (PSQM) score ("perceptual Speech Quality Measurement, PSQM scores", recited in paragraphs

Art Unit: 2616

0006-0009); **regarding claim 19**, wherein an alert threshold level is defined for quality scores of approximately 3.5 ("scores produced by PSQM algorithm of range 1 to 5", recited in paragraphs 0007-0009); **regarding claim 20**, wherein an alert threshold level is defined for quality scores of approximately 3.3 to 3.7 ("scores produced by PSQM algorithm of range 1 to 5", recited in paragraphs 0007-0009); **regarding claim 21**, wherein an alert threshold level is defined for quality scores of approximately 2.8 ("scores produced by PSQM algorithm of range 1 to 5", recited in paragraphs 0007-0009); **regarding claim 22**, wherein an alert threshold level is defined for quality scores of approximately 2.5 to 3.1 ("scores produced by PSQM algorithm of range 1 to 5", recited in paragraphs 0007-0009); **regarding claim 23**, wherein a low level alert threshold level is defined for quality scores of approximately 2.8 ("scores produced by PSQM algorithm of range 1 to 5", recited in paragraphs 0007-0009), and wherein a higher level alert threshold level is defined for quality scores of approximately 3.5 ("scores produced by PSQM algorithm of range 1 to 5", recited in paragraphs 0007-0009 "measures of the quality of the recorded audio stream using PSQM", recited in paragraph 0037); **regarding claims 17, 45**, wherein the quality formula is developed by matching observed quality for communications over the IP network (fig. 1, VoIP Network 10, recited in paragraph 0026) to a standard quality measurement scale ("scores produced by the PSQM algorithm range from a scale of 1 to 5", recited in paragraph 0007), and equating the observed quality to the quality score as the at least two metrics are varied ("the analyzer measure of the quality of the audio stream using quality indicators (jitter, packet loss)",

recited in paragraph 0037, lines 3-9); **regarding claim 46**, the quality formula is designed to approximate a Perceptual Speech Quality Measurement (PSQM) score ("scores produced by PSQM algorithm", recited in paragraph 0007), and wherein a low level alert threshold level is defined for quality scores of approximately 2.8 ("scores produced by the PSQM algorithm range from a scale of 1 to 5", recited in paragraph 0007), and wherein a higher level alert threshold level is defined for quality scores of approximately 3.5 ("scores produced by the PSQM algorithm range from a scale of 1 to 5", recited in paragraph 0007). Therefore, it would have been obvious to one skill in the art at the time the invention was made to modify the features Houh et al. with MeLampy et al. by using features as taught by Baj et al. in order to test the QoS of a component (See paragraph 0011 for motivation).

Houh, MeLampy, Mitsumori, and Baj et al. disclose all the claimed limitation with the exception of being silent with regard to the following claimed features: **regarding claims 23-26**, the color codes, green quality level, yellow quality level, and red quality level; **regarding claims 47-49**, the color codes, green quality level, yellow quality level, and red quality level.

Hussain et al. in a similar field of endeavor discloses the above claimed features: **regarding claims 23-26**, the color codes, green quality level, yellow quality level, and red quality level (see, color marking scheme of green, yellow and red", recited in paragraph 0065); **regarding claims 47-49**, the color codes("the three color marking scheme as red , green, and yellow", recited in paragraph 0037, lines 1-20), green quality level, yellow quality level, and red

Art Unit: 2616

quality level ("the three color marking scheme as red , green, and yellow", recited in paragraph 0037, lines 1-20).

11. **Claim 52** is rejected under 35 U.S.C. 103(a) as being unpatentable over Houh et al (US 2002/0015387 A1) in view of MeLampy et al (US 7,151,781 B2) as applied to claim 31 above, and further in view of Adam et al (US 6,990,071 B2).

Regarding claim 52, Houh et al. discloses the method for near real time quality analysis ("analysis of RTP streams", recited in paragraph 0014, lines 1-9), the packet latency metric ("packet latency as a parameter", recited in paragraph 0049, lines 10-13).

Houh et al and MeLampy et al. disclose all the claimed limitation with the exception of being silent with regard to the following features: wherein the packet latency metric is modeled in the quality formula as an overall formula multiplier.

Adam et al. in a similar field of endeavor discloses the above claimed features: wherein the packet latency metric is modeled in the quality formula as an overall formula multiplier ("multiplying by a function, the latency", recited in col. 10, lines 27-39). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the features of Houh et al. with MeLampy et al. by using features as taught by Adam et al. in order to provide congestion control by rate limiting packet transmissions (See Col. 2, lines 41-51 for motivation).

12. **Claim 53** is rejected under 35 U.S.C. 103(a) as being unpatentable over Houh et al (2002/0015387 A1) in view of MeLampy et al (US 7,151,781 B2) as applied to claim 31 above, and further in view of Clark et al (US 7,075,981 B1).

Houh et al. and MeLampy et al. disclose all the claimed limitation with the exception of being silent with respect to the following claimed features: wherein a packet jitter metric is modeled in the quality formula as either a logarithmic term, or a hyperbolic tangent function or as a piecewise linear function.

Clark et al in the same field of endeavor discloses the claimed features: wherein a packet jitter metric ("measurement of end to end transmission delay", recited in col. 4, lines 48-51) is modeled in the quality formula as either a as a piecewise linear function ("the delay quality degradation is determined using the piecewise linear approximation", recited in col. 5, lines 21-45). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the features of Houh et al. with MeLampy et al. by using features as taught by Clark et al. in order to provide estimation of perceptual quality of multimedia communications (See Col. 3, lines 36-47 for motivation).

13. **Claims 28-30** are rejected under 35 U.S.C. 103(a) as being unpatentable over Houh et al (US 2002/0015387 A1) in view of MeLampy et al (US 7,151,781 B2) in further view of Mitsumori et al (US 6,850,525 B2), Hussain et al (US 2003/0223361 A1).

Regarding claim 28, Houh et al. discloses a near real time quality analyzer (fig. 5, Packet Capture and Passive Analysis Tool 170, recited in paragraph 0060, lines 1-11), comprising: a passive stream collector that passively samples packets (fig. 5, Packets and Analysis Tool 170, "passively analysis on a pair of ports", recited in paragraph 0060) from a stream of Real Time Protocol (RTP) Internet Protocol (IP) packets ("RTP Streams", recited in paragraph 0060, lines 6-16) entering and leaving a switch (fig. 1, Switch Fabric 35, recited in paragraph 0031, lines 1-6), wherein the stream of packets ("received packets from RTP streams", recited in paragraph 0015, lines 9-11) represents a communication session ("RTP streams between the gateways", recited in paragraph 00014, lines 1-7) ("packetized switched applications such as streaming video, video conferencing" recited in paragraph 0037, lines 5-8) between a pair of end points (fig. 5, VoIP Gateways 130, 150, recited in paragraph 0044) carrying analog signals ("packetized switched applications such as streaming video, video conferencing" recited in paragraph 0037, lines 5-8) being transmitted over a transmission path (fig. 5, Transmission path between the VoIP gateways) in an IP network (fig. 2, Packet Network 140, recited in paragraph 0040), and determines ("monitor of a group of RTP streams to determine packet jitter, packet loss", recited in paragraph 0065, lines 1-7) in near real time at least two metrics from the sampled packets (fig. 5, Packet Captures and Analysis Tool 170, "monitors of RTP streams for packet lost, jitter", recited in paragraph 0060, lines 13-19) for the communication session ("RTP streams between the gateways", recited in paragraph 00014, lines 1-7); wherein the at

least two metrics ("packet jitter, number of packet loss form monitored RTP streams", recited in paragraph 0061, lines 6-12) are derived from data contained within Real Time Control Protocol packets ("RTP streams in the packet stream", recited in paragraph 0014, lines 1-7) comprise: at least one metric that measures a quantity of lost packets ("number of packet lost across all streams", recited in paragraph 0061, lines 6-12); and at least one metric that measures a characteristic of packet timing ("packets delay", recited in paragraph 0046, lines 6-12), selected from the group consisting of packet jitter ("packet jitter", recited in paragraph 0015, lines 1-9), packet latency ("packet latency", recited in paragraph 0015, lines 1-9).

Regarding claim 29, Houh et al. discloses the near real time quality analyzer (fig. 5, Packet Capture and Passive Analysis Tool 170, recited in paragraph 0060, lines 1-11) according to claim 28, wherein the stream quality analyzer (fig. 5, Packet Capture and Passive Analysis Tool 170, recited in paragraph 0060, lines 1-11) receives additional metrics ("voice data provided to the test unit 180 to account for delay, jitter, packet loss", recited in paragraph 0047, line 10-15) from network devices ("VoIP gateways 130, 150", recited in paragraph 0044) residing in the IP network (fig. 2, Packet Network 140, recited in paragraph 0040) along the transmission path (fig. 5, Transmission path between the VoIP gateways).

Regarding claim 30, the near real time quality analyzer (fig. 5, Packet Capture and Passive Analysis Tool 170, recited in paragraph 0060, lines 1-11).

Houh et al. discloses all the claimed limitation with the exception of being silent with respect to the following claimed features: **regarding claim 28**, a stream quality analyzer that receives the at least two metrics and calculates a quality score in near real time using a quality formula that combines the at least two metrics, wherein the stream quality analyzer aggregates a plurality of quality scores; a database, receiving the quality score from the stream quality analyzer and storing the quality score indexed to the pair of end points; means for comparing the quality score with a quality threshold and generating an alarm whenever the quality score falls below a quality threshold; and a display that displays the quality score along with historical quality scores associated with the end points; **regarding claim 29**, and wherein the quality formula incorporates functions of the additional metrics; **regarding claim 30**, wherein: a low level alert threshold and a high level alert threshold are established, and wherein, quality scores exceeding the low level alert threshold level but not the higher level alert threshold are color coded as yellow quality level, and wherein quality scores exceeding the higher level alert threshold level are color coded as red quality level, and wherein quality scores lower than the low level alert threshold is color coded as a green quality level.

McLampy et al (US 7,151,781 B2) in the same field of endeavor discloses the above claimed features: **regarding claim 28**, a stream quality analyze (fig. 3, Flow Quality Measurement Engine 348, recited in col. 12, lines 8-15, "determining jitter, packet loss", recited in col. 17, lines 1-12) that receives the at least two metrics ("packet loss, packet latency, jitter", recited in col. 16, lines 66

Art Unit: 2616

– col. 17, lines 12) and calculates a quality score (“QoS score dynamically determined”, recited in col. 13, lines 64 – col. 14, lines 12, fig. 4, QoS Score) in near real time (“multimedia packet flows”, recited in col. 16, lines 66 - col. 17, lines 12) using a quality formula that combines (“QoS is computed that summarizes the three statistics into a single value”, recited in col. 17, lines 23-32) the at least two metrics (“packet loss, packet latency, jitter”, recited in col. 16, lines 66 – col. 17, lines 12), wherein the stream quality analyzer (fig. 3, Flow Quality Measurement Engine 348, recited in col. 12, lines 8-15, “determining jitter, packet loss”, recited in col. 17, lines 1-12) aggregates a plurality of quality scores (“QoS is computed that summarizes the three statistics into a single value”, recited in col. 17, lines 23-32); a database (fig. 4, QoS score column 346, “store a QoS score”, recited in col. 13, lines 64 – col. 14, lines 3) receiving the quality score (“QoS score provided by the flow quality measurement engine 348”, recited in col. 14, lines 4-12) from the stream quality analyzer (fig. 3, Flow Quality Measurement Engine 348, recited in col. 12, lines 8-15, “determining jitter, packet loss”, recited in col. 17, lines 1-12) and storing the quality score (“QoS score stored in real database”, recited in col. 14, lines 5-12) indexed to the pair of end points (fig. 2, First Realm and Second Real 252, recited in col. 7, lines 61 – col. 8, lines 2); **regarding claim 29**, and wherein the quality formula incorporates functions of the additional metrics (“measured jitter, packet loss, and latency of multimedia packets”, recited in col. 13, lines 64 - col. 14, lines 12). Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the features Of Houh et al (US 2002/0015387 A1)

by using features as taught by MeLampy et al. in order to provide bandwidth allocation that ensures the quality of service for transmission of multimedia packets (See Col. 7, lines 1-24 for motivation).

Houh et al. and MeLampy et al. disclose all the claimed limitation with the exception of being silent with respect to the claimed features: **regarding claim 28**, and round trip time, means for comparing the quality score with a quality threshold and generating an alarm whenever the quality score falls below a quality threshold; and a display that displays the quality score along with historical quality scores associated with the end points; **regarding claim 29**, and wherein the quality formula incorporates functions of the additional metrics; **regarding claim 30**, wherein: a low level alert threshold and a high level alert threshold are established, and wherein, quality scores exceeding the low level alert threshold level but not the higher level alert threshold are color coded as yellow quality level, and wherein quality scores exceeding the higher level alert threshold level are color coded as red quality level, and wherein quality scores lower than the low level alert threshold is color coded as a green quality level.

Mitsumori et al (US 6,850,525 B2) in the same field of endeavor discloses the above claimed features: **regarding claim 28**, round trip time ("measures of round trip delay", recited in col. 1, lines 66 – col. 2, lines 10); means for comparing the quality score with a quality threshold and generating an alarm ("monitoring device to trigger an alarm", recited in col. 2, lines 49-60) whenever the quality score falls below a quality threshold ("network performance statistics exceeds a preconfigured threshold", recited in col. 2, lines 49-60); and a display

(fig. 3, "graphical User Interface", recited in col. 6, lines 15-35) that displays the quality score along with historical quality scores ("histograms that show distribution of packet loss, jitter, round-trip delay", recited in col. 5, lines 48-62) associated with the end points (fig. 1, VoIP Points of Presence 104a, 104b, recited in col. 3, lines 37-59); **regarding claim 30**, Mitsumori et al. discloses a low level alert threshold ("trigger an alarm when the network performance statistics exceeds pre-configured threshold", recited in col. 2, lines 49-60) and a high level alert threshold are established ("preconfigured threshold", recited in col. 4, lines 52-67), quality scores exceeding the low level alert threshold level but not the higher level alert threshold ("preconfigured threshold", recited in col. 4, lines 52-67), and wherein quality scores exceeding the higher level alert threshold level, and wherein quality scores lower than the low level alert threshold ("preconfigured threshold", recited in col. 4, lines 52-67). Therefore, it would have been obvious to one skill in the art at the time the invention was made to modify the features of Houh et al. with MeLampy et al. by using features as taught by Mitsumori et al. in order to generate network performance statistics for VoIP network (See Col. 1, lines 43-65 for motivation).

Houh et al, MeLampy et al, and Mitsumori et al. disclose all the claimed limitation with the exception of being silent with regard to the following claimed features: **regarding claim 30**, the color quality coded as yellow, red, and green.

However, Hussain et al (US 2003/0223361 A1) in the same field of endeavor discloses the above claimed features: **regarding claim 30**, the color quality coded as yellow, red, and green ("the three color marker", Red, Yellow,

and Green", recited in paragraph 0037, 0065). Therefore, one skilled in the art would be able to use/incorporate the coloring scheme as taught by Hussein et al. into the teaching features of Houh, MeLampy, and Mitsumori et al. by associating a respective color with a threshold alert level as the quality score of VoIP varies across the transmission path. The color coded scheme as described of the instant application is well-known in the art. The motivation for doing so is to make easy for end users to visualize the quality of VoIP signals during the communication session (See paragraph 0006 for motivation).

MeLampy et al (US 7,151,781 B2) in the same field of endeavor discloses the above claimed features: and calculates a quality score ("QoS score dynamically determined", recited in col. 13, lines 64 – col. 14, lines 12) in near real time ("multimedia packet flows", recited in col. 16, lines 66 - col. 17, lines 12) using a quality formula that combines ("QoS is computed that summarizes the three statistics into a single value", recited in col. 17, lines 23-32) the at least two metrics ("packet loss, packet latency, jitter", recited in col. 16, lines 66 – col. 17, lines 12); **regarding claim 13**, wherein the stream quality analyzer (fig. 3, Flow Quality Measurement Engine 348, recited in col. 12, lines 8-15, "determining jitter, packet loss", recited in col. 17, lines 1-12) aggregates a plurality of quality scores ("QoS is computed that summarizes the three statistics into a single value", recited in col. 17, lines 23-32). Therefore, it would have been obvious to one skill in the art at the time the invention was made to modify the features of Klinker et al. by using features as taught by MeLampy et al. in order to provide

bandwidth allocation that ensures the quality of service for transmission of multimedia packets (See Col. 7, lines 1-24 for motivation).

Allowable Subject Matter

14. **Claims 15-16** would be allowable if rewritten to overcome the rejection(s) under 35 U.S.C. 112, 2nd paragraph, set forth in this Office action and to include all of the limitations of the base claim and any intervening claims. Claims 50-51 are objected to as being dependent upon a rejected base claim, but would be allowable if rewritten in independent form including all of the limitations of the base claim and any intervening claims.

Conclusion

15. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Maurer et al (US 6,700,953 B1), and Hardy William Christopher et al (US 2002/0114296 A1).

16. Any inquiry concerning this communication or earlier communications from the examiner should be directed to CANDAL ELPENORD whose telephone number is (571)270-3123. The examiner can normally be reached on Monday through Friday 7:30AM to 5:00PM EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Kwang Bin Yao can be reached on (571) 272-3182. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Art Unit: 2616

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